REMARKS

It is respectfully submitted that the amendments made to the claims herein are neither being presented nor made in response to the citation of any prior art known to the Applicant or the Applicant's attorneys. These claim amendments are further not made for any reason related to any statutory requirements for patentability. These claim amendments are made solely to more completely claim that to which the Applicant is entitled. Applicant's invention should only be considered limited by the claims as they now exist and the equivalents thereof. It is not the Applicant's intent to narrow any claim element by the amendments made herein. It is submitted that no new matter has been added. A marked-up copy of all pending claims after the amendments made herein is attached to this Preliminary Amendment as Exhibit A. A marked-up copy of all amendments to the specification is attached to this Preliminary Amendment as Exhibit B.

In view of the foregoing, Applicant respectfully requests the thorough and complete

examination of this application and earnestly solicits an early notice of allowance.

Respectfully submitted,

JENKENS & GILCHRIST,
A Professional Corporation

Ross T. Robinson

Ross T. Robinson
Reg. No. 47,031

1445 Ross Avenue, Suite 3200
Dallas, Texas 75202-2799
(214) 855-7300
(214) 855-4300 (fax)

Exhibit A

1. (AMENDED) A method [for] of improving speech quality[,] in a communication system[,] comprising a first terminal unit (TRX1) and a second terminal (TRX2), the method comprising:

[which transmits] transmitting speech signals having a first sampling frequency (F_1); [and a second terminal unit (TRX2), which receives]

receiving said speech signals[,]; [and buffers them]

buffering said speech signals in a playout buffer with said first frequency (F_1); [and plays them]

playing out said speech signals with a second frequency (F_2); [said method **characterised** by]

performing a dynamic sample rate conversion of a speech frame comprising N samples on a sample by sample basis, said dynamic sample rate conversion comprising: [the steps of]

creating an LPC-residual[,] comprising N samples[,] derived from said speech frame;

calculating, for each speech frame, whether a sample should be either added or removed from said LPC-residual;

generating a modified LPC-residual comprising at least one of N-1 [or] and N+1 samples, [if] in response to a determination that said calculating so demands; and

synthesising a speech signal from said modified LPC-residual.

2. (AMENDED) The method of claim 1 [**characterised** in that] wherein the creating step comprises performing an LPC-analysis of the speech frame in order to find LPC-parameters of said speech frame.

3. (AMENDED) The method of claim 1 [**characterised** in that] wherein the creating step comprises using already existing LPC-parameters from a speech decoder.

4. (AMENDED) The method of claim 1 [**characterised** in that] wherein the creating step comprises using an existing LPC-residual from a decoder.

5. (AMENDED) The method of [any of the preceding claims]claim 1, [**characterised** in that] wherein the calculating step comprises deciding whether a sample should be added or removed [on basis of] based on at least one of the following inputs:[;]

[-] [the] sample frequencies of the sending (TRX1) and receiving (TRX2) terminal units;

[-] a voice activity detector signal;

[-] a status of the playout buffer; and

[-] an indicator of [the] a beginning of a talkspurt.

6. (AMENDED) The method of [any of the preceding claims 1-4]claim 1, [**characterised** in

that] wherein the generating step comprises:

selecting [the] a position [where] in the LPC residual at which to add or remove a sample; and

[performing said] adding respective removing of said sample.

7. (AMENDED) The method of claim 6 [further **characterised** by] wherein the step of selecting said position is performed arbitrarily.

8. (AMENDED) The method of claim 6 further [**characterised** in that] comprising the step of finding said position [is found by searching] via a search for a segment of the LPC-residual with low energy.

9. (AMENDED) The method of claim 8 [further **characterised** in that] wherein said segment of low energy is found [by using] via a block energy analysis.

10. (AMENDED) The method of claim 8 [further **characterised** in that] wherein said segment of low energy is found [by using] via a sliding window energy analysis.

11. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said position is found [by] using knowledge about [the] a position of a pitch pulse [together with] and knowledge about a time difference between said pitch pulse and [the] a following pitch pulse to select the position [where] at which to add or remove a sample in the LPC-residual.

12. (AMENDED) The method of claim 11 further [**characterised** in that] comprising the step of finding said pitch pulse [is found by searching] via a search for positions in the LPC residual with high energy.

13. (AMENDED) The method of claim 12 [further **characterised** in that] wherein said positions with high energy are found [by using] via a block energy analysis.

14. (AMENDED) The method of claim 12 [further **characterised** in that] wherein said positions with high energy are found [by using] via a sliding window energy analysis.

15. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said adding [of a sample is done by] comprises adding a zero sample.

16. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said adding [of a sample is done by] comprises adding a zero sample and interpolating surrounding samples.

17. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said removing [of a sample is done by] comprises removing a sample from the LPC-residual.

18. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said adding [of a sample is done by] comprises:

adding a sample in [the] a history of the LPC residual; and

increasing a lag pointer [as] so long as the adding is within [the] an LPC residual history.

19. (AMENDED) The method of claim 6 [further **characterised** in that] wherein said removing [of a sample is done by] comprises:

removing a sample in [the] a history of the LPC residual; and

decreasing a lag pointer [as] so long as the removing is within the LPC residual history.

20. (AMENDED) The method of claim 6 wherein the second terminal unit comprises an adaptive and a fixed codebook; and

[the method further **characterised** in that] wherein said adding [of a sample is done by] comprises:

adding a sample in [the] an output from the adaptive codebook;

extending [the] an output from the fixed codebook; and

increasing a lag pointer [as] so long as the adding is within the LPC residual history.

21. (AMENDED) The method of claim 6 wherein the second terminal unit comprises an adaptive and a fixed codebook; and

[the method further **characterised** in that] wherein said removing [of a sample is done by] comprises:

removing a sample in [the] an output from the adaptive codebook;

shortening [the] an output from the fixed codebook; and

decreasing a lag pointer [as] so long as the removing is within the LPC residual history.

22. (AMENDED) The method of claim 6 wherein the second terminal unit comprises a fixed codebook; and

wherein [the method further **characterised** in that] said adding or removing [of a sample is done by] comprises adding or removing a sample in [the] an output from the fixed codebook.

23. (AMENDED) An apparatus for improving speech quality in a communication system comprising a first terminal unit (TRX1) [transmitting] adapted to transmit speech signals and having a first sampling frequency (F_1) and a second terminal unit (TRX2) [buffering] adapted to buffer said speech signals in a playout buffer with said first frequency (F_1) and [playing them out] to play said speech signals out with a second frequency (F_2), said apparatus comprising:
[**characterised** by]

means for performing a dynamic sample rate conversion of a speech frame comprising N samples on a sample by sample basis, wherein said dynamic sample rate conversion [further characterised by] comprises:

means for creating an LPC-residual[,] comprising N samples[,] derived from said speech

frame;

means for calculating for each speech frame whether a sample should be added or removed from said LPC-residual;

means for generating a modified LPC-residual comprising at least one of N-1 [or] and N+l samples[, if] in response to a determination that said calculating so demands; and

means for synthesising a speech signal from said modified LPC-residual.

24. (AMENDED) The apparatus of claim 23 wherein the means for creating [is **characterised** by further comprising] comprises means for performing an LPC-analysis of the speech frame to find [the] LPC-parameters of said speech frame.

25. (AMENDED) The apparatus of claim 23 wherein the means for creating [is **characterised** by further comprising] comprises means for using existing LPC-parameters from a speech decoder.

26. (AMENDED) The apparatus of claim 23 wherein the means for creating [is **characterised** by further comprising] comprises means for using an existing LPC-residual from a decoder.

27. (AMENDED) The apparatus of [any of claims 23-26]claim 23, wherein the means for calculating [is **characterised** by further comprising] comprises means for deciding if a sample should be added or removed on the basis of a function of at least one of the following inputs:

[-] sample frequencies of sending and receiving terminal units;
[-] a voice activity detector signal;
[-] a status of the playout buffer; and
[-] an indicator of [the] a beginning of a talkspurt.

28. (AMENDED) The apparatus of [any of claims 23-27] claim 23, wherein the means for generating [is **characterised** by further comprising] comprises:

means for selecting [the] a position [where] at which to add or remove samples; and
means for performing [said] adding and removing.

29. (AMENDED) The apparatus of claim 28 wherein the means for selecting [is further **characterised** by] comprises means for arbitrarily selecting said position [where] at which to add or remove samples.

30. (AMENDED) The apparatus of claim 28 wherein the means for selecting [is further **characterised** by] comprises means for searching for the segment of the LPC-residual with the lowest energy.

31. (AMENDED) The apparatus of claim 30 wherein the means for searching [is further **characterised** by] comprises means for performing a block energy analysis.

32. (AMENDED) The apparatus of claim 30 wherein the means for searching [is further **characterised** by] comprises means for performing a sliding window energy analysis.

33. (AMENDED) The apparatus of claim 28 wherein the means for selecting [is further **characterised** by] comprises means for using knowledge about [the] a position of a pitch pulse together with knowledge about a time difference between said pitch pulse and [the] a following pitch pulse to select the position [where] at which to add or remove a sample in the LPC-residual.

34. (AMENDED) The apparatus of claim 33 wherein the means for using knowledge about pitch pulses [is further **characterised** by] comprises means for finding the pitch pulses by searching for positions in the LPC residual with high energy.

35. (AMENDED) The apparatus of claim 34 wherein the means for finding pitch pulses [is further **characterised** by] comprises means for performing a block energy analysis.

36. (AMENDED) The apparatus of claim 34 wherein the means for finding pitch pulses [is further **characterised** by] comprises means for performing a sliding window energy analysis.

37. (AMENDED) The apparatus of claim 28 wherein the means for performing adding or removing [is further **characterised** by] comprises means for adding a zero sample.

38. (AMENDED) The apparatus of claim 28 wherein the means for performing adding or removing [is further **characterised** by] comprises means for removing a sample from the LPC-residual.

39. (AMENDED) The apparatus of claim 28 wherein the means for performing adding or removing [is further **characterised** by] comprises means for adding a zero sample and interpolating surrounding samples.

40. (AMENDED) The apparatus of claim 28 wherein the means for performing adding or removing [is further characterised by] comprises:

means for adding a sample in [the] a history of the LPC residual; and

means for increasing a lag pointer [as] so long as the adding is within the LPC residual history.

41. (AMENDED) The apparatus of claim 28 wherein the means for performing adding or removing [is further **characterised** by] comprises:

means for removing a sample in [the] a history of the LPC residual; and

means for decreasing a lag pointer [as] so long as the removing is within the LPC residual history.

42. (AMENDED) The apparatus of claim 28 wherein the second terminal unit comprises:
an adaptive and a fixed codebook;
[the apparatus further **characterised** by]
means for adding a sample in [the] an output from the adaptive codebook;
means for extending [the] an output from the fixed codebook; and
means for increasing a lag pointer [as] so long as the adding is within the LPC residual history.

43. (AMENDED) The apparatus of claim 28 wherein the second terminal unit comprises:
an adaptive and a fixed codebook;
[the apparatus further **characterised** by]
means for removing a sample in [the] an output from the adaptive codebook;
means for removing a sample in [the] an output from the fixed codebook; and
means for decreasing a lag pointer [as] so long as the removing is within the LPC residual history.

44. (AMENDED) The apparatus of claim 28 wherein the second terminal unit comprises:
a fixed codebook; and
[the apparatus further **characterised** by]
means for adding or removing a sample in [the] an output from the fixed codebook.

Exhibit B

Please replace the paragraph at p. 1, lns. 3-7 with the following paragraph:

The present invention relates generally to apparatuses and methods for improving speech quality in e.g. IP-telephony systems. More particularly, the present invention relates to a method and apparatus for reducing audio [artefacts] artifacts due to overrun or underrun in a playout buffer.

Please replace the paragraph at p. 1, lns. 22-29 with the following paragraph:

Currently, most systems do not deal with the problem that the sampling frequency might differ considerably between the sending and the receiving side. One possible solution proposed in, EP-0680033 A2, works on pitch periods. Adding or removing pitch periods in the speech signal achieves a different duration of a speech segment without affecting other speech characteristics other than speed. This proposed solution might be used as an indirect sample rate conversion method.

Please replace the paragraph at p. 2, lns. 1-11 with the following paragraph:

Another solution uses the beginning of talkspurts as an indication to reset the playout buffer to a specified level. The distance, in number of samples, between two consecutive talkspurts is increased if the receiving side is playing faster than the sending side and decreased if the receiving side is playing slower than the sending side. In IP-telephony solutions[,] using the IP/UDP/RTP-protocols (Internet Protocol/User Datagram Protocol/Real Time Protocol)[; the], a marker flag in

the RTP header is used to identify the beginning of a talkspurt. At the beginning of a talkspurt, the playout buffer is set to a suitable size.

Please replace the paragraph at p. 2, lns. 12-20 with the following paragraph:

The solution according to EP-0680033 A2, where pitch periods are removed or inserted, assumes a fixed conversion factor between the receiving and transmitting side. Therefore, it cannot be used in [dynamical] dynamic systems, i.e. where the sampling frequencies varies. Further, it does not solve the problem with underrun or overrun situations, but is instead focused on changing the playback rate of a speech signal stored in compressed form for playback later and at [another] a different speed [compared to when] to that at which it was stored.

Please replace the paragraph at p. 2, lns. 21-28 with the following paragraph:

Using the method of resetting the playout buffer to a certain size causes problems if there are very long talkspurts, e.g. broadcast from one speaker to several listeners. Since the length of a talkspurt is not defined in the beginning of the talkspurt, the size to reset to might be either too small or too large. If it is to small, underrun will occur and if it is too large, unnecessary delay is introduced. [thus] Thus, the problem persists.

Please replace the paragraph at p.2, lns. 29-31 with the following paragraph:

The general problem with the currently known approaches is that they are static and inflexible. [As a conclusion] Therefore, dynamic solutions are required.

Please replace the paragraph at p. 3, lns. 8-13 with the following paragraph:

When sampling frequencies are not controlled, underrun or overrun might occur in the playout buffer at the receiving side, which causes audible [artefacts] artifacts in the speech signal. To avoid said overrun or underrun there is a need for dynamically keeping the playout buffer to an average size, i.e. controlling the fullness of the playout buffer.

Please replace the paragraph at p. 3, lns. 14-16 with the following paragraph:

One object of the present invention is thus to provide a method for reducing audio [artefacts] artifacts in a speech signal due to overrun or underrun in the playout buffer.

Please replace the paragraph at p. 3, lns. 17-18 with the following paragraph:

Another object of the invention is to dynamically control the fullness of the playout buffer so as not to introduce extra delay.

Please replace the paragraph at p. 3, lns. 19-29 with the following paragraph:

The above mentioned and other objects are achieved by means of dynamic sample rate and conversion of speech frames, i.e. converting speech frames comprising N samples to instead comprise either N+1 or N-1 samples. More specifically, the invention works on an LPC-residual

of the speech frame. [and by] By adding or removing a sample in the LPC-residual, a sample rate conversion will be achieved. The LPC- residual is the output from an LPC-filter, which removes the short-term correlation from the speech signal. The LPC-filter is a linear predictive coding filter where each sample is predicted as a linear combination of previous samples.

Please replace the paragraph at p. 3, lns. 30-33 through p. 4 lns. 1-4 with the following paragraph:
By using the proposed sample rate conversion method, the playout buffer, of e.g. an IP-telephony terminal, can be continuously controlled with only small audio [artefacts] artifacts. Since the method works on a sample-by-sample basis, the playout buffer can be kept to a minimum and hence no extra delay is introduced. The solution also has very low complexity, especially when the LPC-residual already is available, [which] as is the case in e.g. a speech decoder.

Please replace the paragraph at p. 4, lns. 10-13 with the following paragraph:
Although aspects of the invention [has] have been summarised above, the method and [arrangement] apparatus according to the appended claims [independent claims 1 and 23] define the scope of the invention. [Various embodiments are further defined in the dependent claims 2-12 and 24-44.]

Please replace the paragraph at p. 5, lns. 5-20 with the following paragraph:
[The present invention describes, referring] Referring to [figure] FIG. 1, a method for improving speech quality in a communication system [comprising] includes a first terminal unit TRX1

transmitting speech signals having a first sample frequency F_1 and a second terminal unit TRX2 receiving said speech signals, buffering them in a playout buffer 100 with said first frequency F_1 and playing out from said playout buffer with a second frequency F_2. When the buffering frequency F_1 is larger than the playout frequency F_2, the playout buffer 100 will eventually be filled with samples and subsequent samples will have to be discarded. When the buffering frequency F_1 is lower than the playout frequency F_2 the playout buffer will run into starvation, i.e. it will no longer have any samples to play on the output. These two problems are called overrun and underrun, respectively, and [causes] <u>cause</u> audible [artefacts] <u>artifacts</u> like popping and clicking sounds in the speech signal.

Please replace the paragraph at p. 5, lns. 21-24 with the following paragraph:

The above <u>and other</u> problems with underrun and overrun are solved by using dynamic sample rate conversion based on modifying the LPC-residual of the speech signal and will be further described with reference to [figures] <u>FIGS.</u> 2-8.

Please replace the paragraph at p. 6, lns. 6-14 with the following paragraph:

By feeding a speech frame through the LPC-filter, H(z), the LPC-residual is found. The LPC-residual, shown in [figure] <u>FIG</u>. 3, contains pitch pulses P generated by the vocal cords. The distance L between two pitch pulses P is called lag. The pitch pulses P are also predictable, and since they represent the long-term correlation of the speech signal they are predicted through an LTP-filter given by the distance L between the pitch pulses P and the gain b of a pitch pulse P.

The LTP-filter is usually denoted:

Please replace the paragraph at p. 6, lns. 16-19 with the following paragraph:

When the LPC-residual is fed through the inverse of the LTP-filter *F(z)*, an LTP-residual is created. In the LTP-residual, the long-term correlation in the LPC-residual is removed, giving the LTP-residual a noise-like appearance.

Please replace the paragraph at p. 6, lns. 20-27 through p. 7, lns. 1-7 with the following paragraph:

The solution according to the invention modifies the LPC-residual, shown in [figure] FIG. 3, on a sample-by-sample basis. That is, an LPC-residual block comprising N samples is converted to an LPC-residual block comprising either N+l or N-1 samples. The LPC-residual contains less information and less energy compared to the speech signal, but the pitch pulses *P* are still easy to locate. When modifying the LPC-residual, samples [being] that are close to a pitch pulse *P* should be avoided, because these samples contain more information and thus have a large influence on the speech synthesis. The LTP-residual is not as suitable as the LPC-residual to use for modification since the pitch pulse positions *P* are no longer available. [As a conclusion] Thus, the LPC-residual is better suited for modification both compared to the speech signal and the LTP-residual, since the pitch pulses *P* are easily located in the LPC-residual.

Please replace the paragraph at p. 7, lns. 8-9 with the following paragraph:

[The proposed] A sample rate conversion consists of four modules, shown in [figure] FIG. 4:

Please replace the paragraph at p. 7, lns. 12-13 with the following paragraph:

2) LPC-Residual Extraction (LRE) modules 410 that are used to obtain the LPC-residual r_{LPC};

Please replace the paragraph at p. 7, lns. 14-18 with the following paragraph:

3) Sample Rate Conversion Methods (RCM) modules 420 that find the position [where] at which to add or remove samples and determine how to perform the insertion and deletion, i.e. converting the LPC residual block r_{LPC} comprising N samples to a modified LPC-residual block r'_{LPC} comprising N+l or N-1 samples; and

Please replace the paragraph at p. 7, lns. 21-23 with the following paragraph:

[The] An idea behind embodiments of the invention is that it is possible to change the playout rate of the playout buffer 440 by removing or adding samples in the LPC-residual r_{LPC}.

Please replace the paragraph at p. 7, lns. 24-27 through p. 8, lns. 1-11 with the following paragraph:

The SRC module 400 decides whether samples should be added or removed in the LPC residual r_{LPC}. This is done on the basis of at least one of the four following parameters[;]: the sampling

frequencies of the sending TRX1 and receiving terminal units TRx2, information about the speech signal e.g. a voice activity detector signal, status of the playout buffer, [or] an indicator of the beginning of a talkspurt. [These inputs are named] The four parameters are designated SRC Inputs in [the figure] FIG. 4. On the basis of a function of one or several of these parameters the SRC 400 [forms a decision on] decides when to insert or remove a sample in the LPC residual r_{LPC} and optionally which RCM 420 to use. Since digital processing of speech signals usually is made on a frame-by-frame basis, the decision [on] of when to remove or add samples basically is to decide within which LPC-residual r_{LPC} frame the RCM 420 [shall insert or remove] is to insert or remove a sample.

Please replace the paragraph at p. 8, lns. 12-17 with the following paragraph:

There are basically three methods of obtaining the LPC-residual r_{LPC} that is needed as input to the RCM's 420. The methods depend on the implementation of the speech encoder and will be described with reference to [figures] FIGS. 5A-5F. The LRE solution also directly influences the SSM solution, which will become apparent below.

Please replace the paragraph at p. 8, lns. 19-34 through p. 9, lns. 1-4 with the following paragraph:

In [figure] FIG. 5A [is] an analysis-by-synthesis speech encoder 500 with LTP-filter 540 is shown. This is a hybrid encoder where the vocal tract is described with an LPC-filter 550 and the vocal

cords is described with an LTP-filter 540, while the LTP-residual $\hat{r}_{LPC}(n)$ is waveform-compared with a set of more or less stochastic codebook vectors from [the] a fixed codebook 530. The input signal S is divided into frames 510 with a typical length of 10-30 ms. For each frame [an] the LPC-filter 550 is calculated through an LPC-analysis 520 and the LPC-filter 550 is included in a closed loop to find the parameters of the LTP-filter 540. The speech decoder 580 is included in the encoder and consists of the fixed codebook 530, [which] whose output $\tilde{r}_{LTP}(n)$ is connected to the LTP-filter 540, [which] whose output $\hat{r}_{LPC}(n)$ is connected to the LPC-filter 550, [generating] which generates an estimate $\hat{s}(n)$ of the original speech signal $s(n)$. Each estimated signal $\hat{s}(n)$ is compared with the original speech signal $s(n)$ and a difference signal $e(n)$ is calculated. The difference signal $e(n)$ is then weighted by an error-weighting block 560 to calculate a perceptual weighted error measure $e_w(n)$. The set of parameters that gives the least perceptual weighted error measure $e_w(n)$ is transmitted to [the] a receiving side 570.

Please replace the paragraph at p. 9, lns. 6-12 with the following paragraph:

As can be seen in [figure] FIG. 5C, the LPC-residual $\hat{r}_{LPC}(n)$ is the output from the LTP-filter 540. [The] SRC/RCM modules 545 can [thus] be connected directly to [that] the output of the LTP-filler 540 and integrated into the speech encoder. [The] An LRE consists of the fixed codebook 530 and the long-term predictor 540 and the SSM consists of an LPC-filter 550, thus the LRE-module and the SSM-module are natural parts of the speech decoder.

Please replace the paragraph at p. 9, lns. 13-27 with the following paragraph:

If the speech encoder, on the other hand, is an analysis-by-synthesis speech encoder where the LTP-filter 540 is exchanged to an adaptive codebook 590 as shown in [figure] FIG. 5B, the LPCresidual LPC(n) is the output from the sum of the adaptive and the fixed [codebook] codebooks 590 and 530. All other elements have the same function as in [figure] FIG. 5A, [showing the] which shows an analysis-by-synthesis speech encoder with LTP-filter 500. As can be seen in [figure] FIG. 5D the LPC residual $\hat{r}_{LPC}(n)$ is the sum of the output from the adaptive and fixed codebook 590 and 530. The SRC/RCM modules 545 can thus again be connected directly to that output and integrated into the speech encoder as shown in [figure] FIG. 5D. The LRE consists of the adaptive and the fixed codebook 590 and 530 and the SSM consists of an LPC-filter 550, thus the LRE module and the SSM module are again natural parts of the speech decoder.

Please replace the paragraph at p. 9, lns. 28-33 through p. 10, lns. 1-4 with the following paragraph:

When the speech encoder has some sort of backward adaptation, it is not feasible to make alterations in the LPC-residual since this would affect the adaptation process in a detrimental way. In [figure] FIG. 5E is shown how in these cases the parameters $\hat{s}(n)$ from the LPC-filter 550 [could] can be fed to an inverse LPC-filter 525 placed after the speech decoder. After the sample rate conversion has been made in the SRC/RCM modules 545 an LPC-filtering 550 is performed

to reproduce the speech signal. The LRE module consists of the inverse LPC-filter 525 and the SSM module consists of the LPC-filter 550.

Please replace the paragraph at p. 10, lns. 5-15 with the following paragraph:

In [figure] FIG. 5F it is shown how it is possible to produce an LPC residual $\hat{r}_{LPC}(n)$ through a full LPC analysis. The output $\hat{s}(n)$ from the speech decoder is fed to both an LPC analysis block 520 and an LPC-inverse filter 525. After the sample rate conversion has been made in the SRC/RCM modules 545, an LPC filtering 550 is performed to reproduce the speech signal. The LRE consists in this case of the LPC analysis 520 respective the LPC inverse filter 525 and the SSM module consists of the LPC filter 550. Performing an LPC analysis is considered to be well known to a person skilled in the art and is therefore not discussed any further.

Please replace the paragraph at p. 10, lns. 16-23 with the following paragraph:

Referring again to [figure] FIG. 4, assume that the SRC-module 400 has decided that a sample should be added or removed in the LPC residual r_{LPC} and that the LRE module 410 has produced an LPC residual r_{LPC}. The RCM-module 420 then has to find the exact position in the LPC-residual r_{LPC} where to add or remove a sample and performing the adding respective removing. There are four different methods for the RCM-module 420 to find the insertion or deletion point.

Please replace the paragraph at p. 10, lns. 24-28 with the following paragraph:

The first and most primitive method arbitrarily removes or adds a sample whenever this becomes necessary. If the sample rate difference between the terminals is small this will only lead to minor [artefacts] artifacts since the adding or removing is performed very seldom.

Please replace the paragraph at p. 11, lns. 17-26 with the following paragraph:

The fourth method, [as] illustrated in [figure] FIG. 6, uses knowledge about the position P of a pitch pulse, and the lag L between two pitch pulses. With this knowledge [about that], it is possible to calculate a position P' having low energy [and where] at which it is therefore appropriate to add or remove a sample. The new position P' can be expressed as $P' = P+k\cdot L$ [where] , wherein the constant k is selected so that P' is selected to be somewhere in the middle between two pitch pulses, thus avoiding positions with high energy. A typical value of k is in the range of 0.5 to 0.8.

Please replace the paragraph at p. 11, lns. 27-31 with the following paragraph:

When the RCM-module 420 has calculated the position [where] at which to add or remove a sample it must be determined how to perform the insertion or deletion. There are three methods of performing such insertion or deletion depending on the type of LRE-module used.

Please replace the paragraph at p. 12, lns. 1-7 with the following paragraph:

In the first method, either zeros are added or samples with small amplitudes are removed. This

method can be used for all LRE [solution] solutions described above[,]. [see figures 5C-5F.] (See FIGS. 5C-5F.) Notice that in [figures] FIGS. 5C and 5D the SRC/RCM-modules are placed before the synthesis filter SSM, but after the feed back of the LPC residual to the LTP-filter 540 respective the adaptive codebook 590.

Please replace the paragraph at p. 12, lns. 8-15 with the following paragraph:

In the second method, insertion is carried out by adding zeros and interpolating surrounding samples. Deletion is performed by removing samples and preferably smoothing surrounding samples. This method can also be used for all of the LRE solutions described above[,]. [see figures 5C-5F.] See FIGS. 5C-5F. Notice that in [figure] FIG. 5C and 5D the SRC/RCM-modules are placed before the synthesis filter SSM, but after the feed back of the LPC residual to the LTP-filter 540 respective the adaptive codebook 590.

Please replace the paragraph at p. 12, lns. 16-25 with the following paragraph:

In the third method, the SRC/RCM-modules 545 are placed within the feedback loop of the speech decoder[, see figures 5G-5J,] instead of after the feedback loop as in the previous methods. (See FIGS. 5G-5J.) Placing the SRC/RCM-modules within the feedback loop uses real LPC residual samples for the sample rate conversion, by changing the number of components in the LPC-residual. The implementation differs depending on whether it is an analysis-by-synthesis speech encoder with LTP filter shown in [figure] FIG. 5A or an analysis-by-synthesis speech

encoder with adaptive codebook shown in [figure] FIG. 5B[,] that is used.

Please replace the paragraph at p. 12, lns. 26-33 through p. 13, lns. 1-2 with the following paragraph:

For the speech decoder with LTP filter[,] (see [figure 5A] FIG 5A)[,] the SRC/RCM-modules 545 can be placed within the feedback loop in two different ways, either within the LTP feedback loop as shown in [figure] FIG. 5G or in the output from the fixed codebook 530 as shown in [figure] FIG. 5H. For the speech decoder with adaptive codebook[,] (see [figure 5B] FIG. 5B)[,] the SRC/RCM can also be placed in two different ways, i.e. either before[,] ([figure 5J] FIG. 5J)[,] or after, [figure] FIG. 5I, the summation of the outputs from the adaptive and the fixed codebook.

Please replace the paragraph at p. 13, lns. 3-21 with the following paragraph:

The alterations on the LPC residual consists of removing or adding samples just as before, but since the SRC/RCM-modules 545 are placed within the LTP feedback loop, some modifications must be done. The extending or shortening of a segment can be done in three ways either at the respective ends of the segment or somewhere in the middle of the segment. Figure 7 shows the case where the LPC residual is extended by copying two overlapping segments, segment 1 and segment 2, from the history of the LPC residual to create the longer LPC residual. The normal case when no insertion or deletion is needed would be to copy N samples. Shortening the LPC residual is achieved by copying two segments that has a gap between them instead of being

overlapped. As before, it is important that a pitch pulse is not doubled or removed since this would introduce perceptual [artefacts] artifacts. Hence, an analysis should be performed in order to evaluate where to add or remove segments. This analysis is preferably made by using the same methods as discussed above regarding how to find the position where to add or remove a sample in the RCM-module.

Please replace the paragraph at p. 14, lns. 11-16 with the following paragraph:

[The] Embodiments of the invention being thus described, it will be obvious that the same may be varied in many ways. Such variations are not to be regarded as a departure from the scope of the invention, and all such modifications as would be obvious to a person skilled in the art are intended to be included within the scope of the following claims.

Please replace the paragraph at p. 24 lns. 3-20 with the following paragraph:

[The present invention relates to methods for improving speech quality in e.g. an IP-telephony system. The invention reduces audio artefacts being] Audio artifacts due to overrun or underrun in a playout buffer caused by the sampling rates at a sending and receiving side not being at the same rate are reduced. [The inventive solution modifies an] An LPC-residual is modified on a sample-by-sample basis. The LPC-residual block [comprising] , which includes N samples, is converted to a block comprising N+1 or N-1 samples. A sample rate controller [400] decides whether samples should be added to or removed from the LPC-residual. The exact position [where] at which to add respective remove samples is either chosen arbitrarily or found by

searching for low energy segments in the LPC-residual. A speech synthesiser module [430] then reproduces the speech. By using the proposed sample rate conversion method the playout buffer [440] can be continuously controlled. Furthermore, since the method works on a sample-by-sample basis the buffer can be kept to a minimum and hence no extra delay is introduced.

[(Publication figure: Figure 4)]